In the Claims:

Please amend the Claims as shown below:

1. (Canceled).

2. (New) A method for processing a musical signal, comprising simulating a cabinet simulation model responsive to a virtual sampling rate.

3. (New) The method of Claim 2, where the cabinet simulation model comprises at least a finite impulse response filter represented as

$H(z) = a_o + a_1 z^{-m} + a_2 z^{-2m} + \ldots + a_L z^{-LM}$, where

M is 1 / the virtual sampling rate, and L is a number of taps in the finite impulse response filter.

4. (New) The method of claim 3, further comprising

selecting a sampling value, and

determining the virtual sampling rate responsive to the selected sampling value.

5. (New) The method of Claim 2, further comprising storing the virtual sampling rate in a memory.

6. (New) A method for processing a musical signal, comprising:

warping between a first simulation model and a second amplification simulation model; and

producing a generated simulation model.

7. (New) The method of Claim 6, where the first simulation model, the second simulation model and the generated simulation model all comprise at least one of an amplifier simulation model, a cabinet simulation model, a reverb simulation model, a time-variant effect simulation model such as a modulation effects simulation model

including at least one of a chorus modulation effect, a flanger modulation effect, a phaser modulation effect, a pitch-shifter modulation effect, a rotary simulator modulation effect, and an intelligent harmony modulation effect, and a delays simulation model.

8. (New) A digital signal processor for processing a musical signal, comprising:

a cabinet simulation model operating responsive to a virtual sampling rate.

9. (New) The digital signal processor of Claim 8, where the cabinet comprises at least a finite impulse response filter represented as

$H(z) = a_o + a_1 z^{-m} + a_2 z^{-2m} + \ldots + a_L z^{-LM}$, where

M is 1 / virtual sampling rate, and L is a number of taps in the finite impulse response filter.

10. (New) The digital signal processor of Claim 8, further comprising a selector coupled to the cabinet modeler allowing selection of a selector value used in the determination of the virtual sampling rate value.

11. (New) The digital signal processor of Claim 8, further comprising a memory coupled to the cabinet modeler for storing the virtual sampling rate.

12. (New) A system for processing a musical signal, comprising:

a first simulation model;

a second simulation model; and

a simulation model generator coupled with the first and second amplification simulation models, the simulation model generator capable of warping between the first and second simulation models, and the simulation model generator capable of producing a generated simulation model.

13. (New) The method of Claim 12, where the first simulation model, the second simulation model and the generated simulation model all comprise at least one of an amplifier simulation model, a cabinet simulation model, a reverb simulation model, a time-variant effect simulation model such as a modulation effects simulation model including at least one of a chorus modulation effect, a flanger modulation effect, a phaser modulation effect, a pitch-shifter modulation effect, a rotary simulator modulation effect, and an intelligent harmony modulation effect, and a delays simulation model.